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Laurence, deceased et al.

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[45] **Date of Patent: Feb. 26, 1991**

[54] **HEARING AIDS**

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[73] Assignee: **National Research Development Corporation,** London, England

[21] Appl. No.: **465,523**

[22] Filed: **Jan. 17, 1990**

Related U.S. Application Data

[63] Continuation of Ser. No. 72,475, Jul. 13, 1987, abandoned.

[30] **Foreign Application Priority Data**

Jul. 11, 1986 [GB] United Kingdom 8616959

[51] **Int. Cl.**[5] **H04R 25/00**
[52] **U.S. Cl.** **381/68.4;** 381/107
[58] **Field of Search** 381/68.4, 68, 94, 104, 381/106, 107, 108, 68.2

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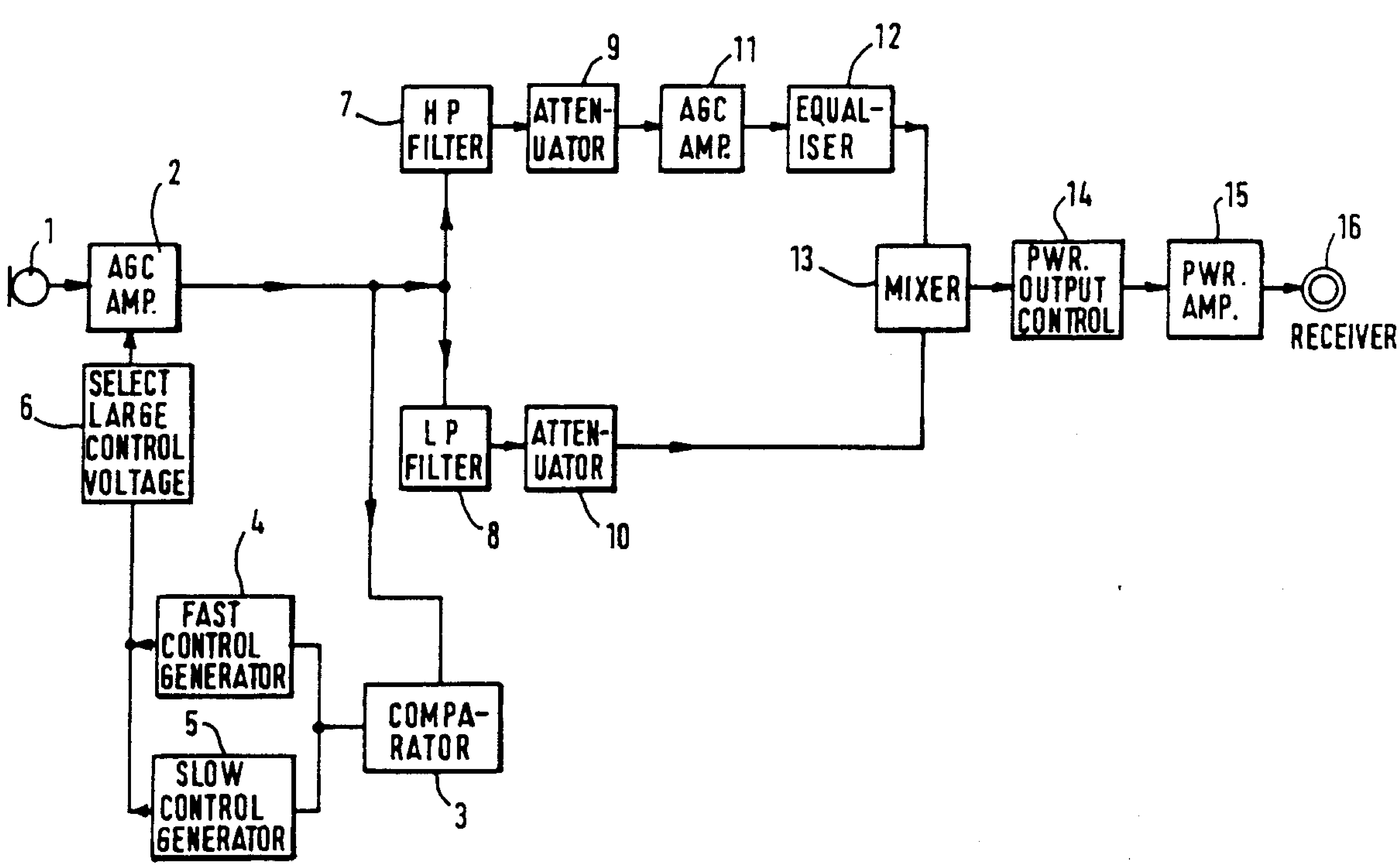
Primary Examiner—Forester W. Isen
Attorney, Agent, or Firm—Cushman, Darby & Cushman

[57] **ABSTRACT**

A hearing aid comprising an automatic gain control arrangement which provides an output signal of substantially constant average level. The automatic gain control arrangement has a slow control generator which provides a relatively slow release time and adapts the gain to the average level of the input signal, and a fast generator which provides a relatively fast release time to allow impulsive sounds or transient signals to be accommodated. When a sufficiently loud impulsive sound is received, the fast generator takes control of the gain to avoid discomfort or overloading. The automatic gain control arrangement limits the dynamic range for sound pressure levels above a threshold, such as 60 dB SPL.

21 Claims, 8 Drawing Sheets



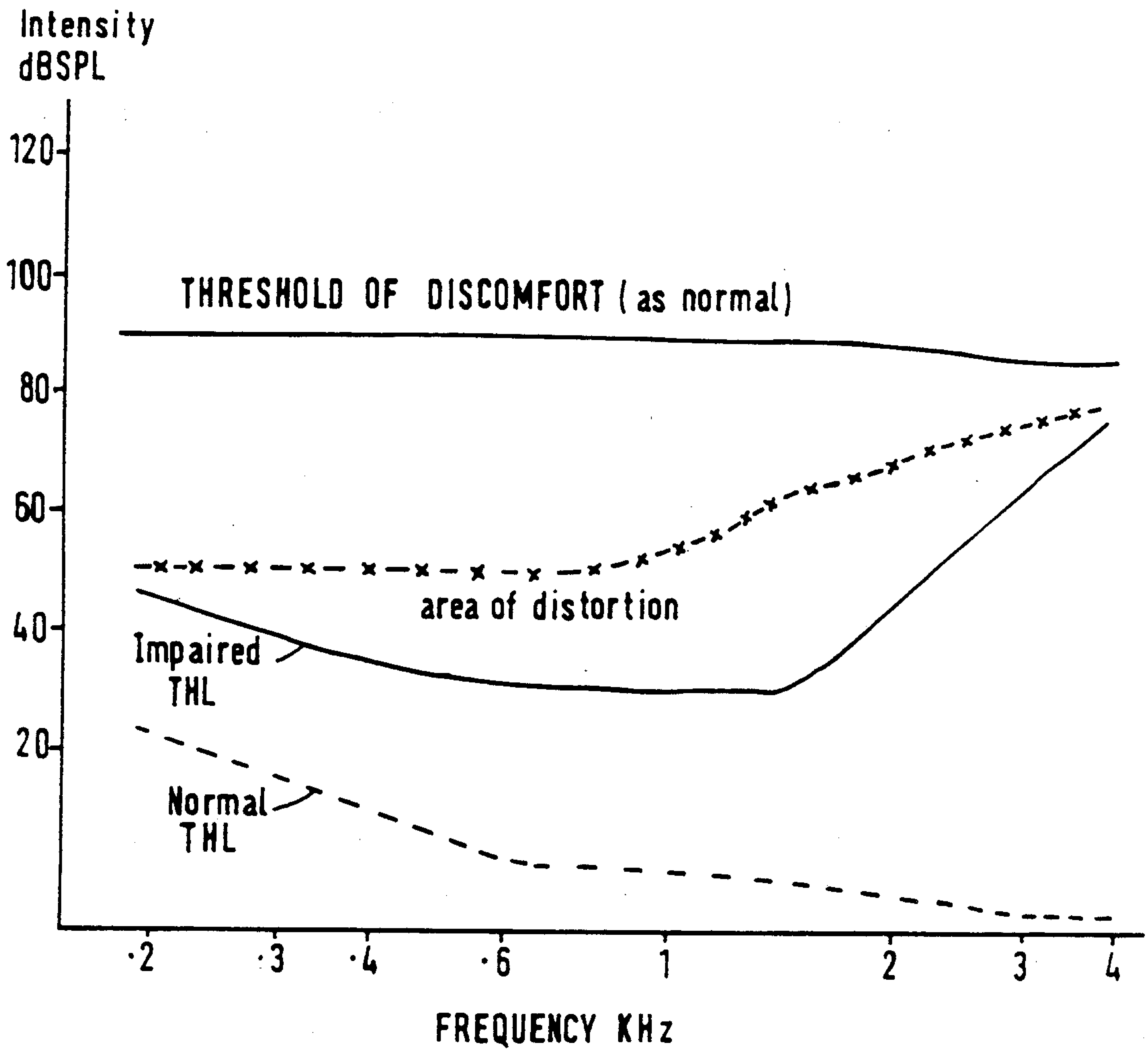

Intensity
dBSPL
120
100
80
60
40
20
THRESHOLD OF DISCOMFORT (as normal)
area of distortion
Impaired
THL
Normal
THL
·2
·3
·4
·6
1
2
3
4
FREQUENCY KHz

FIG.1.

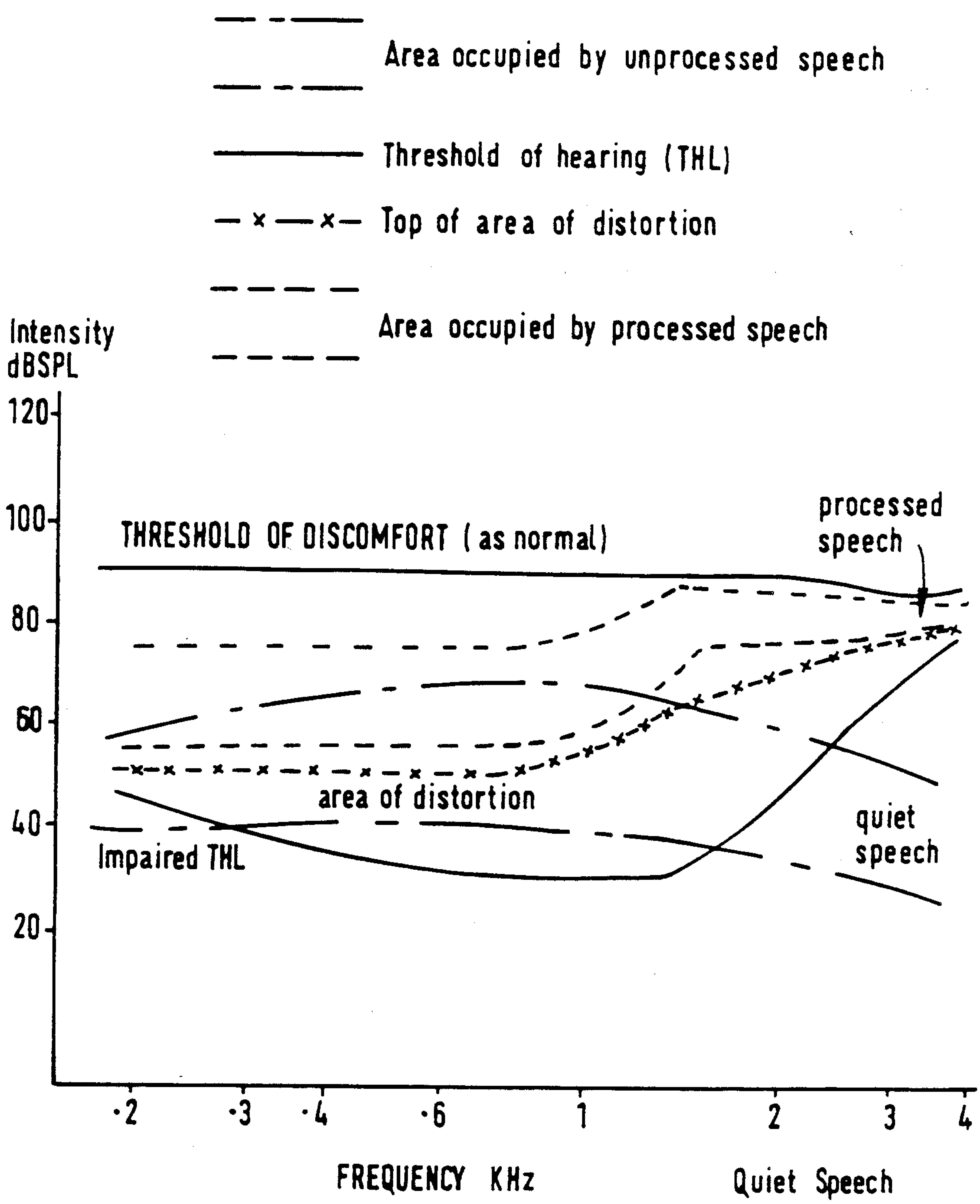

Area occupied by unprocessed speech
Threshold of hearing (THL)
Top of area of distortion
Area occupied by processed speech
Intensity dBSPL
120
100
80
60
40
20
THRESHOLD OF DISCOMFORT (as normal)
processed speech
area of distortion
quiet speech
Impaired THL
·2
·3
·4
·6
1
2
3
4
FREQUENCY KHz
Quiet Speech

FIG. 2.

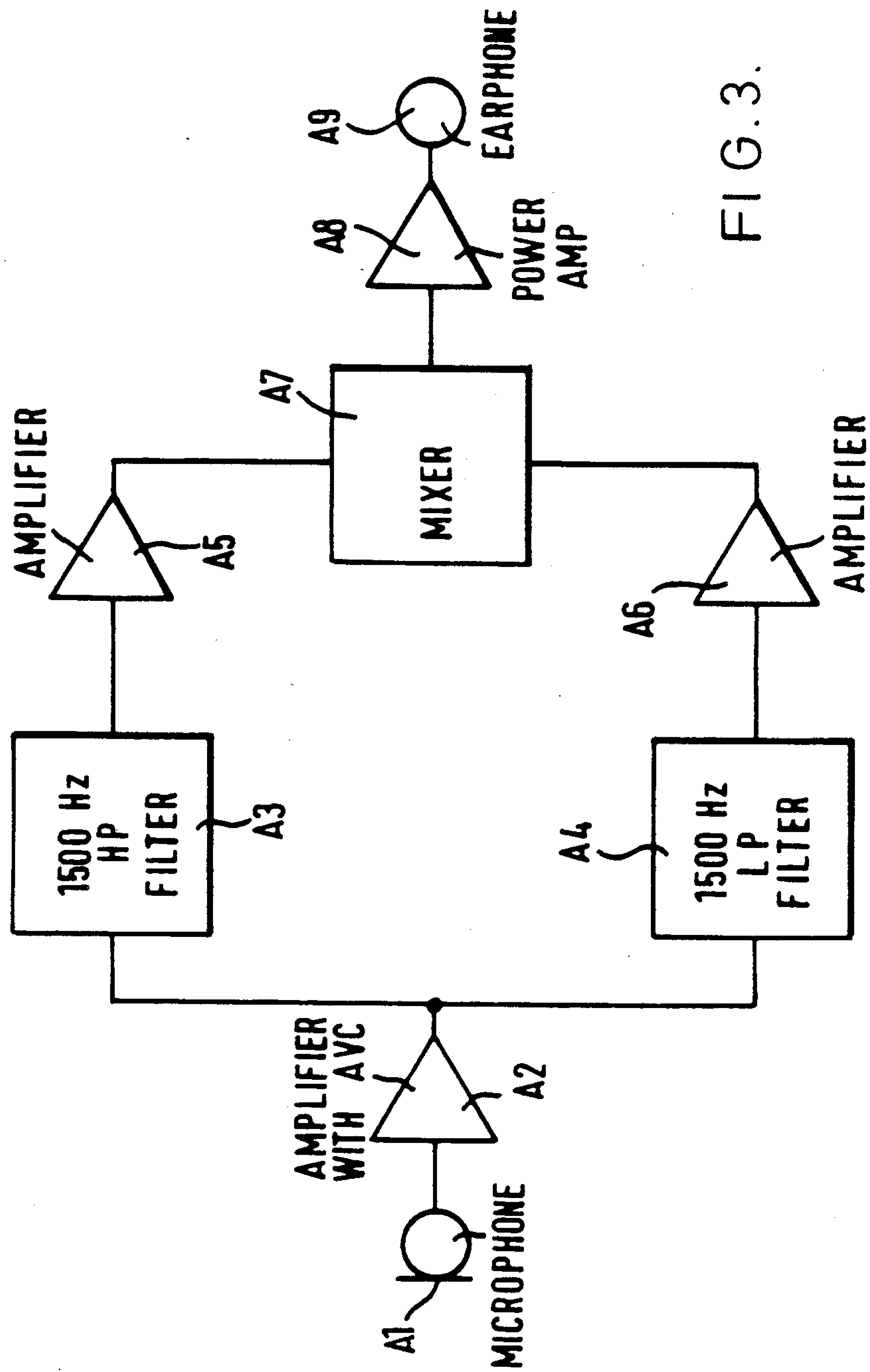

MICROPHONE
A1
AMPLIFIER WITH AVC
A2
1500 Hz HP FILTER
A3
1500 Hz LP FILTER
A4
AMPLIFIER
A5
AMPLIFIER
A6
MIXER
A7
POWER AMP
A8
EARPHONE
A9

FIG. 3.

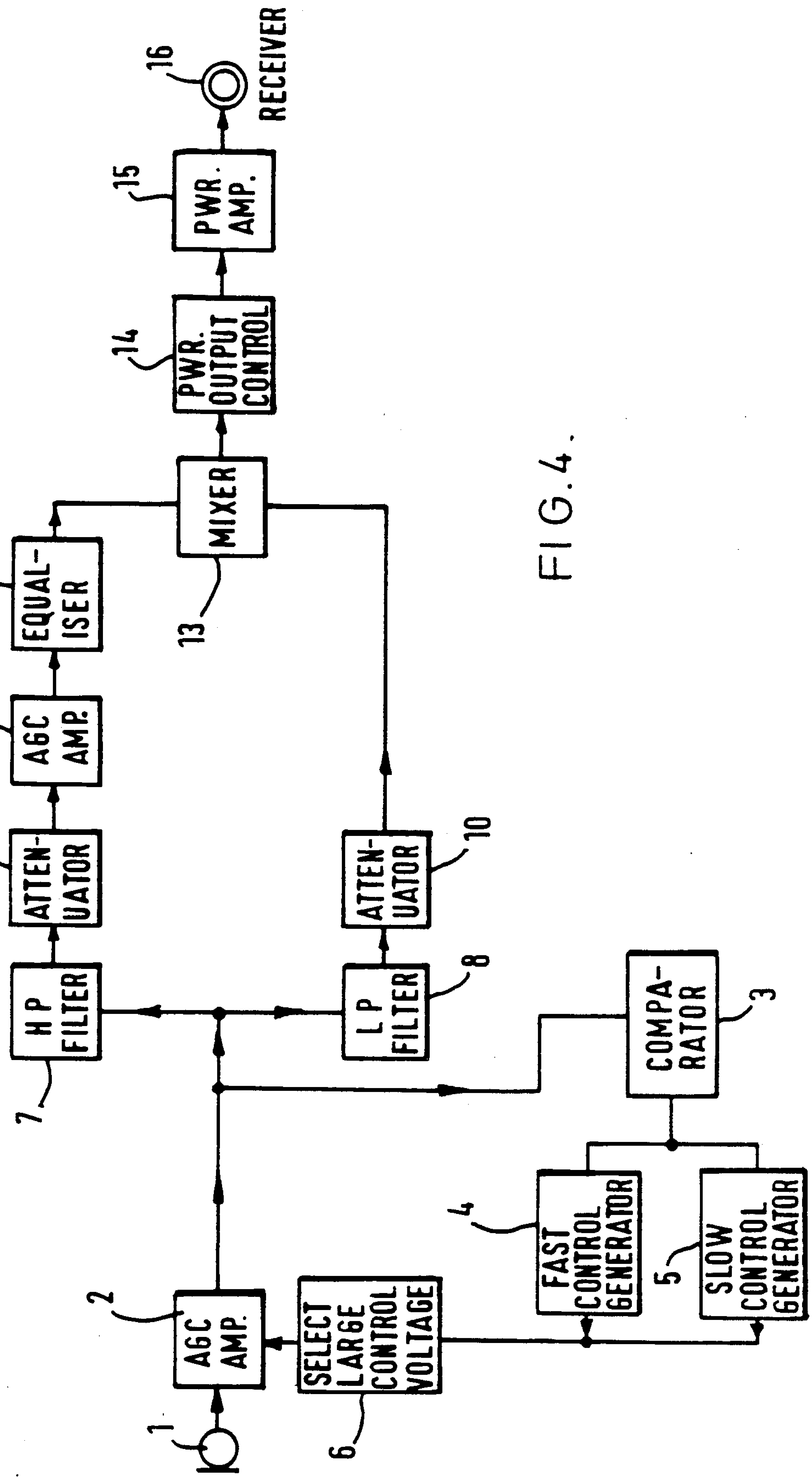
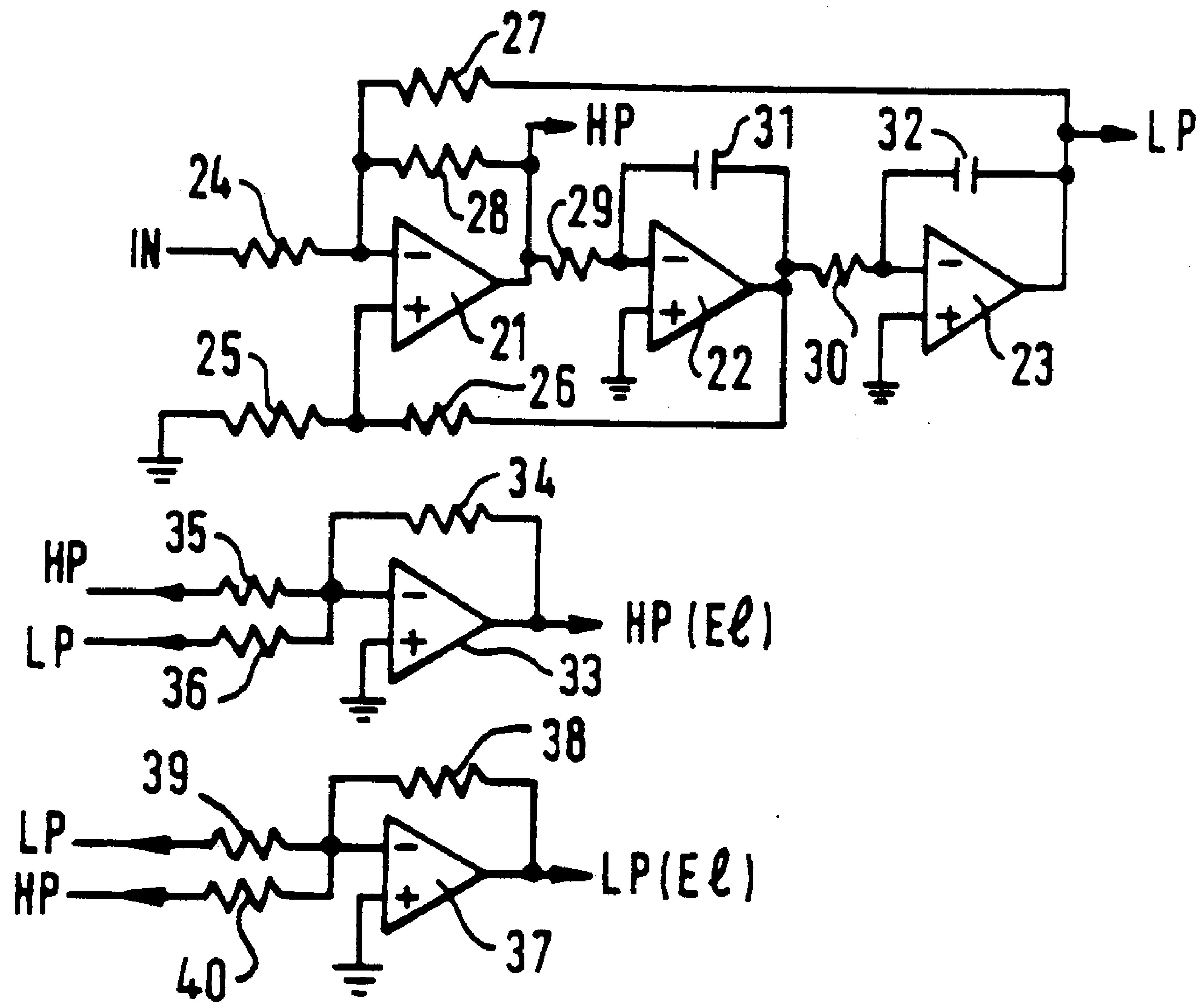
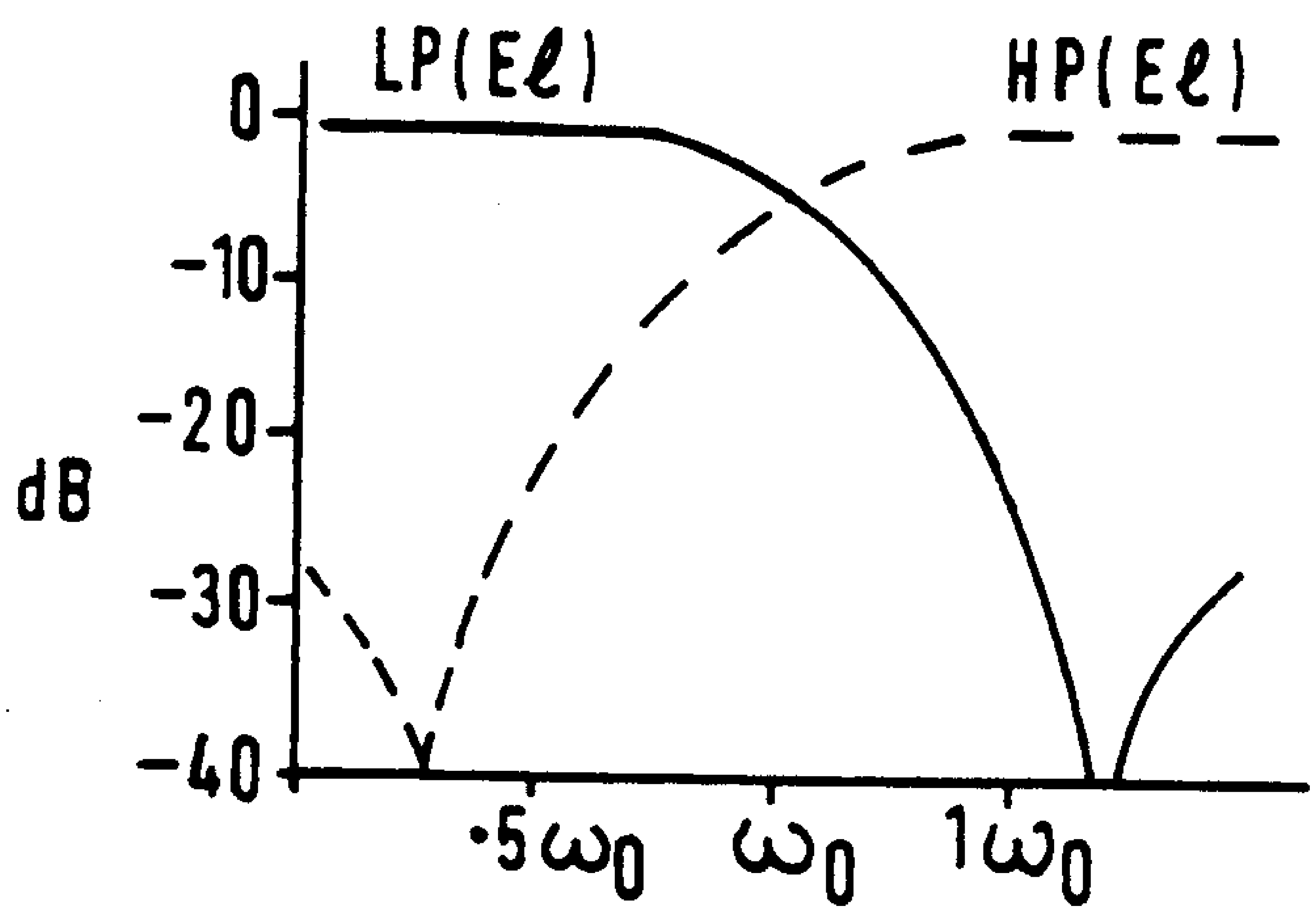

RECEIVER
16
15
PWR. AMP.
14
PWR. OUTPUT CONTROL
MIXER
13
12
EQUAL-ISER
11
AGC AMP.
9
ATTEN-UATOR
10
ATTEN-UATOR
HP FILTER
7
LP FILTER
8
COMPA-RATOR
3
4
FAST CONTROL GENERATOR
5
SLOW CONTROL GENERATOR
2
AGC AMP.
SELECT LARGE CONTROL VOLTAGE
6
1

FIG. 4.



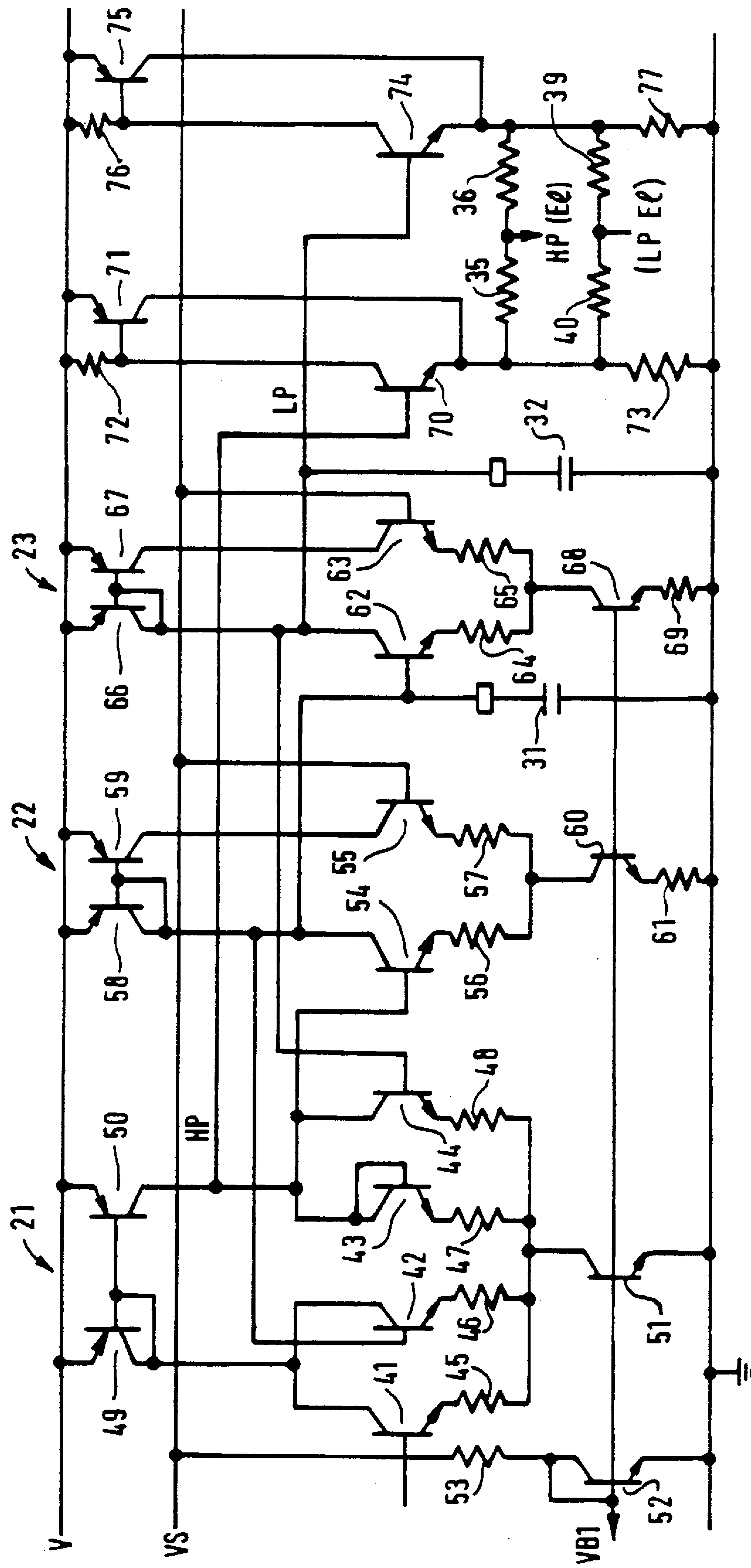
27
HP
31
32
LP
24
28
29
IN
21
22
30
23
25
26
34
35
HP
LP
HP (Eℓ)
36
33
38
39
LP
HP
LP(Eℓ)
40
37

FIG. 5.



LP(Eℓ)
HP(Eℓ)
0
-10
-20
dB
-30
-40
·5ω0
ω0
1ω0

FIG. 6.


V
VS
HP
LP
VB1
HP (Eℓ)
(LP Eℓ)
21
22
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32
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42
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FIG. 7.

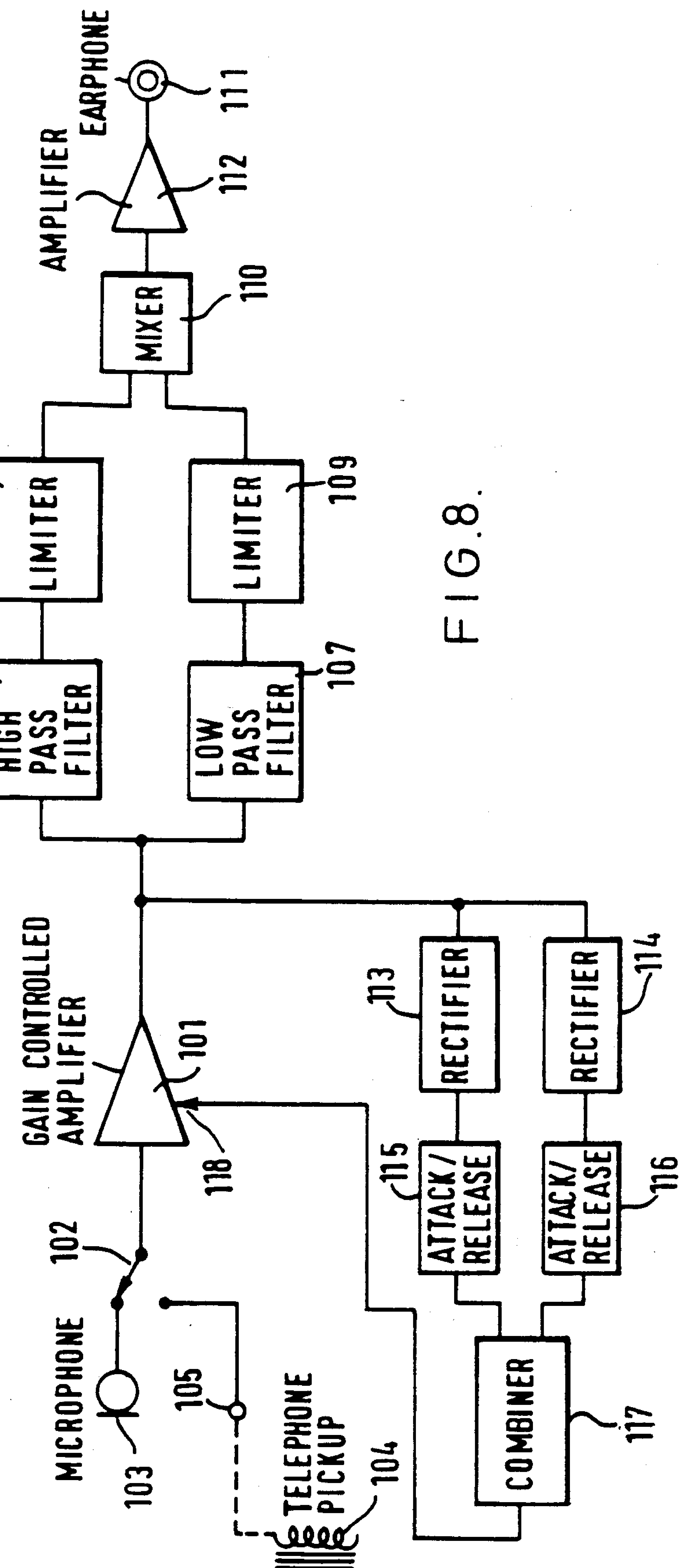
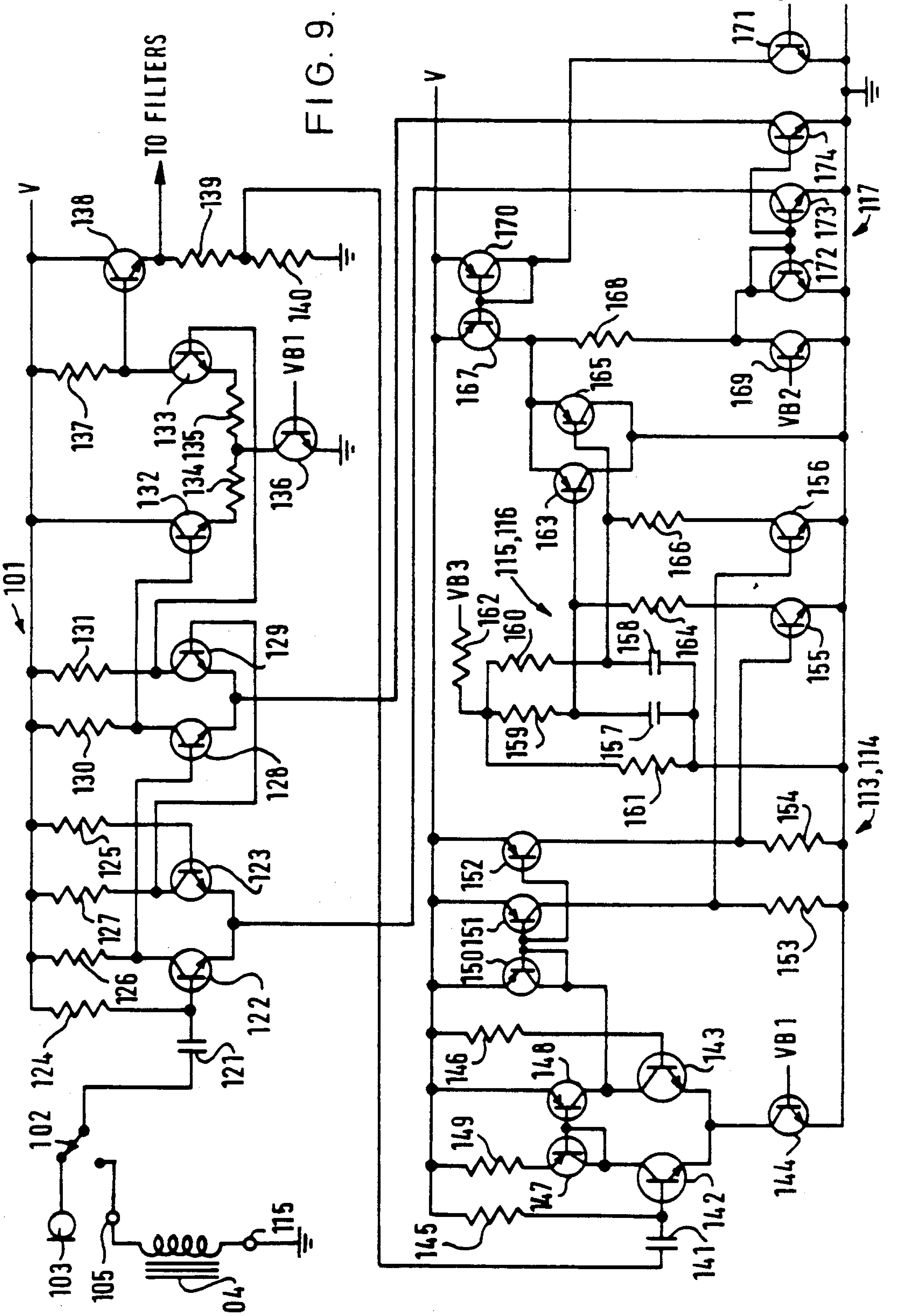

EARPHONE
111
AMPLIFIER
112
MIXER
110
108
LIMITER
LIMITER
109
106
HIGH PASS FILTER
LOW PASS FILTER
107
GAIN CONTROLLED AMPLIFIER
101
118
RECTIFIER
113
RECTIFIER
114
ATTACK/ RELEASE
115
ATTACK/ RELEASE
116
COMBINER
117
102
MICROPHONE
103
105
TELEPHONE PICKUP
104

FIG.8.

46552

8



TO FILTERS
V
VB1
VB2
VB3
101
113,114
115,116
117
102
103
104
105
115
121
122
123
124
125
126
127
128
129
130
131
132
133
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FIG. 9.

HEARING AIDS

This is a continuation of application Ser. No. 07/072,475, filed July 13, 1987, now abandoned.

BACKGROUND OF THE INVENTION

(a) Field of the Invention

The present invention relates to hearing aids, and more particularly, to a hearing aid for delivering acoustic signals of a substantially constant average signal level to persons with moderate to severe hearing loss.

Hearing aids are apparatuses, normally electronic, for amplifying sound and supplying it to the ear of a person whose hearing is impaired. A microphone or other transducer converts sounds, particularly speech, or other signals into an electrical signal which is then amplified and supplied to an earphone inserted in the outer ear of the patient.

It is very common for a hearing impaired person to find that a hearing aid helps him to understand speech from one person in a quiet room but is useless if others are talking at the same time. The reason for this is that a patient who has reduced sensitivity of hearing, perhaps due to sensori neural deafness, usually has other deficiencies as well which are not normally measured.

The parameters important for speech understanding, particularly in noise, include:

Sensitivity: if the average hearing loss at 500, 1000 and 2000 Hz is less than 35 dB a patient can manage well unaided in quiet situations. If the loss is greater, he will benefit from amplification of the weaker speech components;

Recruitment: this is common, particularly with sensori neural deafness, which accounts for 80% of all deafness. The threshold of hearing is elevated. Below it the patient hears nothing. Above it the patient hears with the intensity of a normal ear so that sounds are either inaudible or loud;

Frequency discrimination: a healthy ear can detect a 1% change in frequency of a tone. A bad ear may only detect a 20% change, or even hear noise rather than a tone;

Frequency selectivity: the peripheral auditory system analyzes the incoming complex sounds of speech into their component frequencies by what are called auditory filters. If the ability to do this is impaired, speech recognition, particularly in noise, becomes very difficult. Understanding speech depends on comparing moment to moment changes in the spectrum of the speech sound;

Temporal discrimination: normal ears can perceive gaps in speech as small as 3 mS. Impaired ears may not perceive gaps of 10 mS. For speech recognition small gaps must be recognised. Noise makes matters worse by filling in the gaps; and

Temporal masking: if a weak sound follows after a loud sound it may not be heard. This effect is often worse with impaired ears and causes weak consonants after loud vowels to be lost. Noise makes matters worse by masking weak consonants even further.

Not all these difficulties may be present in any particular patient, but a patient with an audiogram showing reduced sensitivity often has deficiences in some of these parameters as well.

A peculiarity of speech is that the vowels are loud and of low frequency and consonants are weak and of high frequency. Another peculiarity of speech is that most of the information is in the consonants.

FIG. 1 of the accompanying drawings, which is a graph of sound level against frequency, shows the threshold of hearing (THL) and the threshold of discomfort of a sensori-neural deaf patient. The threshold of discomfort is the same as for normal hearing whereas the threshold of hearing is raised compared with a normal THL. The temporal and frequency parameters which are needed to perceive speech tend to be better above an area which is labelled area of distortion, so that speech should be presented to the patient in the area above the area of distortion.

FIG. 2 of the accompanying drawings, which is a graph of sound level against frequency, shows the long term spectral distribution of ordinary quiet speech as the area between chain dotted lines to ensure that all components are heard by the patient, the speech envelope should be processed so that it fits between the broken lines. This means that high frequencies should be amplified more than low frequencies and that the dynamic range, particularly at high frequencies, should be reduced. Low frequencies must be prevented from becoming too loud or they will mask high frequencies.

(b) Description of the Prior Art

One previous but unsuccessful way of trying to achieve this is to provide a tone control to make the amplification greater at high frequencies, and then use automatic volume control (AVC) to reduce the dynamic range. This does not work because all the energy in speech is in the low pitched vowels and most of the information is in the weak high pitched consonants. AVC causes amplifier gain to drop when a loud vowel occurs so that a following weak consonant is suppressed and is not heard.

FIG. 3 of the accompanying drawings is a block circuit diagram of another known hearing aid of the type disclosed in EP 0077688. Signals from a microphone A1 are supplied to another amplifier A2. The amplifier A2 has conventional AVC which only operates at high speech levels above 70 dB SPL and is used to prevent very loud speech from overloading the system. The attack and release time constants are 2 milliseconds (mS) and 300 mS, respectively. The speech signal is then split into an upper frequency band above 1500 Hz and a lower band below 1500 Hz by high pass and low pass filters A3 and A4, respectively. The degree of compression needed by the patient is set in amplifier A5 for high frequencies to suit his dynamic range. Low frequency vowels do not go through this channel. The amplifier A5 limits the dynamic range of the signals with attack and release time constants of 2 mS and 10 mS, respectively. Similarly, the appropriate compression is set in a limiting amplifier A6 for low frequencies having attack and release time constants of 2 mS and 30 mS, respectively. More compression than necessary is used here so that loud sounds do not mask high frequencies. A mixer A7 combines the outputs of the amplifiers A5 and A6 and supplied the combined signal to a power amplifier A8 whose gain is adjusted to suit the threshold of discomfort of the patient. A receiver or earphone A9 delivers speech to the patient's ear.

In such a hearing aid, the amplifier A2 starts compressing or limiting at a fairly high threshold. Also, a relatively simple AVC system is used with attack and release time constants which are a compromise between conflicting requirements. Namely, first of all the input

AGC amplifier should present all speech signals to the rest of the hearing aid at the same average level. Otherwise, the high frequency AGC channel will recover between speech peaks when high intensity speech is presented to the microphone and the hearing aid will sound noisy. Secondly the input AVC amplifier should extract the true average value of the incoming speech. The hearing aid shown in FIG. 3 uses input AVC with an attack time of 2 mS, so that the patient is protected against loud noises like a bang of a door, and a 300 mS release time to 90% of full sensitivity. To extract a real average, the release time should be much longer, for instance about 5 S, but then the bang of a door would disable the aid for an obtrusive time.

SUMMARY OF THE INVENTION

According to the invention, there is provided a hearing aid comprising: automatic gain control means for compressing the dynamic range of an input signal so as to provide an output signal of substantially constant average signal level in response to input signal levels within a predetermined range, the automatic gain control means comprising first and second gain control means for controlling gain in response to the input signal, the first gain control means having a first release time and the second gain control means having a second release time, the first release time being longer than the second release time; a plurality of signal processing channels for providing processed output signals, each of the signal processing channels receiving the output signal from the automatic gain control means and including frequency dependent filtering means; and means for providing a combined output signal from the processed output signals and for supplying the combined output signal to an output transducer.

By providing two gain control means with different release times, it is possible to control the gain automatically so as to accommodate two different input signals. The first gain control means with the longer release time can be used to provide gain control in response to the average input signal level, whereas the second gain control means with the shorter release time allows the gain to be reduced in response to transient or impulsive signals without suppressing following lower level signals. For this purpose, the release time of the first gain control means may be greater than 500 mS, preferably greater than or substantially equal to one second, for instance substantially equal to two seconds or even five seconds. The release time of the second gain control means is preferably less than 100 mS. For instance, it may be less than 75 mS to such as 50 mS.

Preferably, the attack time of the first gain control means is greater than the attack time of the second gain control means. The attack time of the first gain control means may be less than 100 mS, for instance substantially equal to 50 mS. Preferably, the attack time of the second gain control means is less than 10 mS, for instance substantially equal to 2 mS. The lower limit of the predetermined range may correspond to a sound pressure level of 55 or 60 dB. The upper limit may be 100 dB.

The plurality of signal processing channels may include a first channel comprising a low pass filter for passing a lowest frequency range and presettable gain control means. The or each of the other signal processing channels may include means for compressing or limiting dynamic range.

The hearing aid may include an equalizer for compensating ear canal resonance and/or output transducer frequency response. When an ear canal is occluded with an ear mould containing an ear piece, a standing wave is created which reduces sound pressures above 3 kHz progressively until a reduction of about 10 dB is reached at 4 kHz. Also, the earphone is normally driven by a voltage source in the form of a low output impedance amplifier to which it presents an inductive load. This results in a 6 dB/octave reduction in drive current through the earphone and hence in output level. The equalizer may be used to compensate for either or both of these phenomena. The automatic gain control means may comprise first and second gain control means for reducing gain in response to the level of the input signal, the first gain control means having a longer release time than the second gain control means.

The automatic gain control means may include a single variable gain element which is controlled by the first and second gain control means. Preferably, gate means are provided between the variable gain element and the first and second gain control means so as to pass the signal from the first and second gain control means which provides lower gain. Preferably, the first and second gain control means are arranged to reduce the gain of the amplifier when the level of the input signal exceeds first and second thresholds, respectively, the first threshold being lower than the second threshold. Thus, the gain of the amplifier is normally determined by the average level of the input signal. However, when a peak or transient signal of substantially higher level is received, the amplifier gain is reduced temporarily so as to avoid overloading or discomfort without suppressing or masking the signal following the transient or peak.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be further described, by way of example, with reference to FIGS. 1 to 9 of the accompanying drawings, in which:

FIG. 1 is a graph of sound level against frequency showing the threshold of hearing and the threshold of discomfort of a sensori-neural deaf patient.

FIG. 2 is a graph of sound level against frequency showing the long term spectral distribution of ordinary quiet speech.

FIG. 3 is a block circuit diagram of a hearing aid of the prior art.

FIG. 4 is a block circuit diagram of a hearing aid constituting a first embodiment of the invention.

FIG. 5 is a circuit diagram of filters of the hearing aid of FIG. 4;

FIG. 6 is a graph showing the frequency response of the filters of FIG. 5;

FIG. 7 is a detailed circuit diagram of a practical implementation of the filters of FIG. 5;

FIG. 8 is a block circuit diagram of a hearing aid constituting a second embodiment of the invention; and

FIG. 9 is a detailed circuit diagram of part of the hearing aid of FIG. 8.

DESCRIPTION OF THE PREFERRED EMBODIMENT

The hearing aid shown in FIG. 4 comprises a microphone 1 whose output is connected to the input of a gain controlled or AGC amplifier 2. The output of the amplifier 2 is supplied to a comparator 3 which compares the signals from the amplifier 2 and supplies an output corresponding to the amount by which the input signal exceeds a predetermined level, in particular a sound pressure level of 60 dB. The output of the comparator is supplied to a fast control generator **4** and a slow control generator **5**, whose outputs are connected to a circuit **6** for selecting the larger of the control voltages from the generators **4** and **5**. The output of the circuit **6** is supplied to a gain controlled input of the amplifier **2** so as to provide an automatic gain control function.

The fast control generator **4** has an attack time of 2 mS and a release time of 50 mS so as to protect a patient against initial loud speech syllables or loud transients, such as the banging of a door. The slow generator **5** has an attack time of 50 mS and a long release time, such as five seconds. The slow generator **5** thus extracts the average value of an input speech signal. The circuit **6** passes the larger of the signals from the generators **4** and **5** so that, in the absence of transient or impulsive sounds, the gain of the amplifier **2** is controlled by the average signal level but, when a relatively large transient sound occurs, the gain of the amplifier **2** is reduced quickly and recovers quickly.

The output of the amplifier **2** is also supplied to a high pass filter **7** and a low pass filter **8** provided in high pass and low pass channels, respectively. In the high pass channel, the output of the filter **7** is supplied to an adjustable attenuator **9** whose output is connected to the input of an automatic gain control amplifier **11** for compressing or limiting the dynamic range of the signals passing through the high pass channel. The output of the amplifier **11** is supplied to an equalizer **12** which compensates for sound pressure reduction above 3 kHz caused by standing waves in the ear canal when occluded by an ear mould containing an earphone. The equalizer **12** may also be arranged to compensate for the frequency response of the output transducer, which is normally of the inductive type, and supplied by a low output output impedance amplifier.

The output of the low pass filter **8** is connected to the input of an adjustable attenuator **10**.

The outputs of the equalizer **12** and the attenuator **10** are mixed or summed in a mixer **13**. The output of the mixer is supplied to a power output control **14**, whose output is connected to a power amplifier **15**. The output of the power amplifier **15** drives a receiver shown as an earphone **16**.

The automatic gain control circuit **2-6** compresses the dynamic range of signals whose amplitudes exceed **60** dB SPL so as to perform a limiting function in order to supply signals of substantially constant average level to the filters **7** and **8**. Relatively high frequency signals, for instance above 1500 Hz, are supplied to the high pass channel whereas relatively low frequency signals are supplied to the low pass channel. The high pass channel thus processes signals which are principally generated by consonants and the dynamic range compression and amplification allows the level of the consonants to be boosted so as to lie in the area between the broken lines in FIG. 2. It is thus possible for consonant signals to be placed between the threshold of discomfort and the area of distortion so as to maximize the audibility of consonants and hence improve the intelligibility of speech.

The low pass channel processes signals which are predominantly associated with vowels. As is apparent from the curve defining the impaired threshold of hearing in FIG. 2, the dynamic range of a patient's hearing at relatively low frequencies is not too greatly reduced and, in many cases, it is not therefore necessary to provide any dynamic range compression at the lower frequencies. Instead, the levels of such frequencies may simply be adjusted in order to ensure the maximum intelligibility of speech. Accordingly, in such circumstances, the low pass channel does not require any gain compression, and therefore merely contains the attenuator **10** for setting the balance or relative levels between the signals from the high pass and low pass channels supplied to the mixer **13**.

FIG. **5** shows one form of filter arrangement which may be used as the high pass and low pass filters **7** and **8** in FIG. **4**. This filter arrangement provides combined high pass notch and low pass notch or elliptic filter characteristics as shown in FIG. **6** at HP (E1) and LP (E1), respectively. The filter arrangement comprises a state variable filter having an input IN which receives the signal from the amplifier **2**. The state variable filter includes operational amplifiers **21**, **22**, and **23**, resistors **24** to **30**, and capacitors **31** and **32**, and is of conventional type. In order to provide the high pass and low pass filters **7** and **8**, the high pass output HP and the low pass output LP of the state variable filter are used.

In order to provide the high pass elliptic response, the high pass and low pass outputs of the state variable filter are supplied to a mixer comprising an operational amplifier **33**, negative feedback resistor **34**, and input resistors **35** and **36**. The value of the resistor **36** is made equal to ten times the values of the resistor **35** so that the high pass output of the state variable filter is summed with 1/10th of the low pass output to provide the output signal HP (E1) as illustrated in FIG. **6**.

In order to provide the low pass elliptic characteristic, the low pass and high pass outputs of the state variable filter are supplied to a mixer comprising an operational amplifier **37**, a negative feedback resistor **38**, and input mixing resistors **39** and **40**. The value of the resistor **40** is made ten times the value of the resistor **39** so that the low pass output of the state variable filter is summed with 1/10th of the value of the high pass output to provide the low pass elliptic output signal LP (E1) as illustrated in FIG. **6**.

An advantage of this filter arrangement is that all of the components except the integrating capacitors **31** and **32** may be included on a common integrated circuit. Such an implementation of the filter arrangement is shown in FIG. **7**. In FIGS. **5** and **7**, the like reference numerals refer to like parts. The operational amplifier **21** comprises a differential input stage including transistors **41** to **44** and resistors **45** to **48** provided with a current mirror collector load formed by transistors **49** and **50** and a constant tail current source formed by a current mirror including a transistor **51**, a transistor **52** and a current defining resistor **53**. The input branch of the current mirror supplies a reference voltage VB1, and is connected between a common line and a stabilized voltage line VS derived by stabilizing means (not shown) from a supply line V.

The operational amplifier **22** is formed by a differential stage comprising transistors **54** and **55** with emitter resistors **56** and **57** and a current mirror collector load formed by transistors **58** and **59**. The differential stage has a constant tail current source formed by a transistor **60** whose base is connected to receive the bias voltage VB1 and whose emitter is connected via resistor **61** to the common line. The integrating capacitor **31** is connected between the collector of the transistor **54** and the common line, and the resistor **29** is provided by the output impedance of the transistor **54**.

The operational amplifier **23** is also formed by a differential pair of transistors **62** and **63** provided with emitter resistors **64** and **65**, a current mirror collector load formed by transistors **66** and **67**, and a constant tail current source formed by a transistor **68** and a resistor **69**. The capacitor **32** is connected between the collector of the transistor **62** and the common line, and the resistor **30** is formed by the output impedance of the transistor **62**.

The high pass and low pass outputs HP and LP, respectively, are connected to the inputs of combined mixing means comprising a first compound emitter follower, which is formed by transistors **70** and **71** and resistors **72** and **73**, and a second compound emitter follower comprising transistors **74** and **75** and resistors **76** and **77**. The resistors **35** and **36** are connected in series between the emitters of the transistors **70** and **74**. Similarly, the resistors **39** and **40** are connected in series between the emitters of the transistors **70** and **74**. The connection point between the resistors **35** and **36** and the connection point between the resistors **39** and **40** form the high pass and low pass outputs HP (E1) and LP (E1), respectively, of the filter arrangement.

The hearing aid shown in FIG. **8** comprises a gain-controlled amplifier **101** whose input is connected to a changeover switch **102**. The changeover switch **102** selects an input signal either from a microphone **103** forming part of the hearing aid or from a telephone pickup **104** which is optionally connected to an input terminal **105** of the hearing aid. The output of the amplifier **101** is connected to a high pass filter **106** and a low pass filter **107**. The outputs of the filters **106** and **107** are connected to the inputs of dynamic range processing circuits shown as limiters **108** and **109**, respectively. The outputs of the limiters **108** and **109** are supplied to a mixing circuit **110** whose output drives an earphone **111** via an amplifier **112**. Adjustable attenuators may be provided to allow adjustment of the action of the limiters **108** and **109**, to allow the relative levels of the signals to the mixer **110** to be adjusted, and to allow the overall volume to be set.

The output of the amplifier **101** is also connected to the inputs of rectifiers **113** and **114**. The outputs of the rectifiers **113** and **114** are supplied to the inputs of attack/release circuits **115** and **116**, respectively. The outputs of the circuits **115** and **116** are combined by a combiner or gate **117**, which supplied a control signal to a gain control input **118** of the amplifier **101**.

The attack/release circuit **115** comprises a time constant circuit having an attack time substantially equal to 75 mS and a release time substantially equal to two seconds. The attack/release circuit **116** comprises a time constant circuit having an attack time substantially equal to 2 mS and a release time substantially equal to 75 mS.

An input signal selected by the switch **102** is supplied to the amplifier **101**. The rectifier **113** and circuit **115** provide a signal which corresponds to the average level of the input signal. The rectifier **114** and the circuit **116** provide a signal which corresponds to the level of peak or transient signals of relatively short duration. The combiner circuit **117** passes the signal from the circuits **115** and **116** corresponding to the lower gain of the amplifier **101**. Thus, in the absence of impulsive sounds or other transient signals, the gain of the amplifier **101** is controlled by the average level of the input signal. If a transient input signal is received with a peak level which exceeds by a predetermined amount or proportion the average signal level, the gain of the amplifier **101** is reduced so as to reduce the output level of this transient signal. Once the transient has passed, the gain of the amplifier **101** is restored to the value corresponding to the output signal from the circuit **115** following the release time of the circuit **116** so as to avoid suppression of average level or relatively low level signals following the transient.

FIG. **9** shows in more detail the amplifier **101**, the rectifiers **113** and **114**, the attack/release circuits **115** and **116**, and the combiner **117**. Like reference numerals correspond to like parts. The output of the switch **102** is supplied via a coupling capacitor **121** to the base of a transistor **122** which, together with a transistor **123**, forms a long-tailed pair with controlled tail current. Resistors **124** and **125** are connected between the bases of the transistors **122** and **123**, respectively, and a positive supply line V to provide base bias current. The collectors of the transistors **122** and **123** are connected to the positive supply line V via collector load resistors **126** and **127**, respectively, and are connected to the bases of transistors **128** and **129**, respectively. The transistors **128** and **129** are connected together as a long tail pair with controlled tail current. The collectors of the transistors **128** and **129** are connected to the positive supply line V via collector load resistors **130** and **131**, respectively, and to the bases of transistors **132** and **133**, respectively. The emitters of the transistors **132** and **133** are connected via resistors **134** and **135**, respectively, to another output branch of the current mirror comprising a transistor **136** whose base is connected to the transistor **52**. The collector of the transistor **132** is connected to the positive supply line V and the collector of the transistor **133** is connected to the positive supply line via collector load resistor **137** and to the base of a transistor **138** connected as an emitter-follower with its collector connected to the positive supply line. The emitter of the transistor **138** forms the output of the amplifier **101** and is connected to the filters **106** and **107** shown in FIG. **8**. The emitter is also connected via series-connected resistors **139** and **140** to the common line.

The connection between the resistors **139** and **140** is connected via a coupling capacitor **141** to the base of a transistor **142** which, together with a transistor **143**, comprises a long tailed pair. The emitters of the transistors **142** and **143** are connected to a current source comprising a common emitter-connected transistor **144** whose base receives the bias voltage VB1. The bases of the transistors **142** and **143** are connected via bias resistors **145** and **146**, respectively, to the positive supply line V. The collectors of the transistors **142** and **143** are connected to a current mirror formed by transistors **147** and **148** and a resistor **149** in the emitter circuit of the transistor **147**. The value of the resistor **149** is selected so as to provide a current amplification ratio of 10:1 between the input current flowing through the transistor **147** and the output current flowing through the transistor **148**.

The collectors of the transistors **143** and **148** are connected to the input branch of a current mirror circuit including a transistor **150**. The current mirror circuit has two output branches comprising transistors **151** and **152** whose collectors are connected to the common line via resistors **153** and **154**, respectively. The collectors of the transistors **151** and **152** are also connected to the bases of transistors **155** and **156**, respectively, whose emitters are connected to the common line.

The attack/release circuits **115** and **116** comprise capacitors **157** and **158**, respectively. The capacitors **157** and **158** are connected in series with resistors **159** and **160**, respectively, the two series circuits being connected in parallel between the common line and a connection between resistors **161** and **162**, whose other ends are connected to the common line and a voltage VB3 supplied by a reference voltage source (not shown), respectively.

The connection between the resistor **159** and the capacitor **157** is connected to the base of a transistor **163** and via a resistor **164** to the collector of the transistor **155**. The connection between the resistor **160** and the capacitor **158** is connected to the base of a transistor **165** and via a resistor **166** to the collector of the transistor **156**. The collectors of the transistors **163** and **165** are connected to the common line. The emitters of the transistors **163** and **165** are connected to the collector of a transistor **167** and to one end of a resistor **168** whose other end is connected to the collector of a transistor **169**. The transistor **167** forms the output branch of a current mirror whose input branch is provided by a transistor **170**. The collector and base of the transistor **170** are connected to a constant current source comprising a transistor **171** in the common emitter mode provided with the base bias voltage VB1.

The transistor **169** is connected in the common emitter mode and is provided with a base bias voltage, VB2. The collector of the transistor **169** is connected to the collector and base of a transistor **172** forming the input branch of a current mirror circuit having two output branches formed by the transistors **173** and **174**. The collector of the transistor **173** is connected to the emitters of the transistors **122** and **123** so as to supply the tail current thereof. The collector of the transistor **174** is connected to the emitters of the transistors **128** and **129** so as to supply the tail current thereof.

The gain of the amplifier **101** is controlled by varying the tail currents through the transistors **122**, **123** and **128**, **129**. A proportion of the output signal of the amplifier, determined by the values of the resistors **139** and **140**, is supplied to the rectifier circuits **113** and **114** which also perform a comparison function. When the input signal is below a threshold value, the transistors **142** and **143** hold the transistors **151** and **152** off so that the transistors **155** and **156** are turned off. The capacitors **157** and **158** are therefore charged to the value of the voltage VB3, which is such as to cause the transistors **163** and **165** to pass a total of 5 microamps. The constant current source comprising the transistor **169** passes a constant current of 4 microamps. The current mirror comprising the transistors **167** and **170** reflects a current of 20 microamps supplied by the constant current source including the transistor **171**. Thus, a current of 11 microamps flows through the transistor **172** forming the input branch of a current mirror. A current of 11 microamps therefore flows in the collectors of each of the transistors **173** and **174** and this forms the tail currents of the transistors **122**, **123** and **128**, **129**, respectively. The gain of the amplifier **101** is therefore at a maximum value.

Whenever the input signal exceeds a threshold, for instance when an impulsive sound occurs, the transistors **151** and **152** are turned on. The transistors **155** and **156** are therefore also turned on and the capacitors **157** and **158** are discharged at a rate determined by the resistors **164** and **166**, respectively. If the amplitude of the input signal is sufficiently high, the transistors **163** and **165** pass a total of 14 microamps, so that a current of 2 microamps flows into the transistor **172** and is reflected by the transistors **173** and **174** into the tails of the long tailed pairs comprising the transistors **122**, **123** and **128**, **129**. The gain of the amplifier **101** is thus set to a minimum. The ratio of maximum to minimum gain of the amplifier **101** may, for instance, by approximately equal to 36 dB.

The attack and release time constants of the first gain control circuit **115** are determined principally by the values of the capacitor **157** and the resistors **159** and **164**. The attack and release time constants are set to be substantially equal to 75 mS and two seconds, respectively. This may be achieved by making the capacitor **157** equal to 4.7 microfarads and by making the resistors **159** and **164** equal to 1 megohm and 22 kilohms, respectively. The second gain control circuit **116** is provided with attack and release times of 2 and 75 mS, respectively. In order to achieve this, the capacitor **158** may have a value of 0.1 microfarads and the resistors **160** and **166** may have values of 1 megohm and 22 kilohms, respectively. The resistors **164** and **166** control the attack times whereas the resistors **159** and **160** control the release times. The arrangement is such that whichever signal at the bases of the transistors **163** and **165** corresponds to minimum gain is used to control the gain of the amplifier **101**.

Many of the components shown in FIG. **9**, for instance all the transistors and most of the resistors, may be provided by a monolithically integrated circuit. Preferably, such a circuit is adapted to operate from a relatively low supply voltage, for instance 1.2 volts. Thus, all the electronics, the microphone **103**, the earphone **111**, and a battery may be accommodated within a small housing capable of being worn unobtrusively at or behind the ear of a patient.

The amplifier **101** and associated gain control components provide broadband automatic gain control between the microphone and subsequent frequency dividing and dynamic range processing parts of the hearing aid. The use of the dual AGC arrangement allows the hearing aid to accommodate signals of varying levels including transient signals corresponding to impulsive noises without overloading or causing discomfort and without suppressing quieter sounds following impulsive or transient sounds. A signal of substantially constant average level is thus supplied to the subsequent processing parts.

The filter arrangement shown in FIGS. **5–7** may be used as the filters **106** and **107** of FIG. **8**. Also, the gain control arrangement of FIG. **9** may be used as the automatic gain control **2-6** of FIG. **4**. The arrangement of FIGS. **5–7** and **9** may be provided in a common integrated circuit so as to reduce the space requirement and provide a compact design of the hearing aid.

We claim:

1. A hearing aid comprising:

automatic gain control means for compressing the dynamic range of an input signal by providing a reduced gain when the input signal has an amplitude exceeding a predetermined threshold so as to produce an output signal of substantially constant average signal level in response to input signal levels within a predetermined range, said automatic gain control means further comprising,

first gain control means for controlling gain in response to a normal speech level component of the input signal, said first gain control means having a first release time,

second gain control means for controlling gain in response to a transient or impulsive component of the input signal having a larger amplitude than said normal speech level component, said second gain control means being set to reduce gain in response to said transient or impulsive component relative to said normal speech level component without suppressing low amplitude speech signals, and said second gain control means having a second release time which is shorter than said first release time;

a plurality of signal processing channels for providing processed output signals, each of said signal processing channels receiving the output signal from said automatic gain control means and including frequency dependent filtering means; and

means for providing a combined output signal from the processed output signals and for supplying the combined output signal to an electro-acoustic output transducer.

2. A hearing aid as claimed in claim 1, in which the predetermined range corresponds to a sound pressure level range between 55 and 100 dB.

3. A hearing aid as claimed in claim 1, in which said plurality of signal processing channels includes a first channel comprising a low pass filter for passing a lowest frequency range and presettable gain control means.

4. A hearing aid as claimed in claim 3, in which said plurality of signal processing channels includes at least one further signal processing channel comprising means for reducing dynamic range.

5. A hearing aid as claimed in claim 1, in which said plurality of signal processing channels comprises a state variable filter providing a high pass output for supplying a high pass signal and a low pass output for supplying a low pass signal, each of said high pass signal and low pass signals being input into first and second channels, said frequency dependent filtering means of said first channel comprising a low pass elliptic filter, and said frequency dependent filtering means of said second channel comprising a high pass elliptic filter.

6. A hearing aid as claimed in claim 5, in which said high pass elliptic filter comprises first mixing means for attenuating the low pass signals, and for mixing the high pass signal with the attenuated low pass signal, and in which said low pass elliptic filter comprises second mixing means for attenuating the high pass signal and for mixing the low pass signal with the attenuated high pass signal.

7. A hearing aid as claimed in claim 1, in which the first release time of said first gain control means is greater than 500 mS and the second release time of said second gain control means is less than 100 mS.

8. A hearing aid as claimed in claim 7, in which the first release time of said first gain control means is at least 1 second.

9. A hearing aid as claimed in claim 8, in which the first release time of said first gain control means is at least 2 seconds.

10. A hearing aid as claimed in claim 9, in which the first release time of said first gain control means is substantially equal to 5 seconds.

11. A hearing aid as claimed claim 7, in which the second release time of said second gain control means is less than 75 mS.

12. A hearing aid as claimed in claim 11, in which the second release time of said second gain control means is substantially equal to 50 mS.

13. A hearing aid as claimed in claim 11, in which said first gain control means has a first attack time of less than 100 mS.

14. A hearing aid as claimed in claim 13, in which the first attack time of said first gain control means is not greater than or equal to 50 mS.

15. A hearing aid as claimed claim 11, in which said second gain control means has a second attack time of less than 10 mS.

16. A hearing aid as claimed in claim 15, in which the second attack time of said second gain control means is substantially equal to 2 mS.

17. A hearing aid as claimed in claim 1, in which said first gain control means has an attack time substantially equal to the second release time of said second gain control means.

18. A hearing aid as claimed in claim 1, in which said automatic gain control means includes a single variable gain element controlled by said first and second gain control means.

19. A hearing aid as claimed in claim 18, in which said automatic gain control means includes gate means for controlling said variable gain element to respond to the one of said first and second gain control means signalling lower gain.

20. A hearing aid as claimed in claim 19, in which said first gain control means responds to input signal levels above a first threshold and said second gain control means responds to input signal levels above a second threshold, the first threshold being lower than the second threshold.

21. A hearing aid comprising:

automatic gain control means for compressing the dynamic range of an input signal by providing a reduced gain when the input signal has an amplitude exceeding a predetermined threshold and for producing an output signal of substantially constant average signal level in response to input signal levels within a predetermined range,

said automatic gain control means comprising first and second gain control means for controlling gain response to the input signal,

said first gain control means operating to control gain in response to normal speech levels by responding to signal levels above a first amplitude threshold,

said second gain control means operating to control gain in response to transient or impulsive signals of large amplitude without suppressing low amplitude speech signals by responding to signal levels above a second amplitude threshold,

the first threshold being lower than the second threshold,

said first gain control means having a first release time and said second gain control means having a second release time, the first release time being longer than the second release time;

a plurality of signal processing channels for providing processed output signals, each of said signal processing channels receiving the output signal from said automatic gain control means and including frequency dependent filtering means; and

means for providing a combined output signal from the processed output signals and for supplying the combined output signal to an electro-acoustic output transducer.

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